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Sir:

Transmitted herewith for filing under 37 C.F.R. §1.53(b) is the patent application of
Inventor(s): Shoutarou YODA

For: SPEECH RECOGNITION SYSTEM AND METHOD

XX Specification and Claims (42 pages)
XX 10 sheets of drawings
XX Newly executed Declaration and Power of Attorney
XX Return Receipt Postcard
XX An assignment of the invention to Pioneer Corporation with accompanying PTO-1595 Form
XX A certified copy of Japanese Application(s) No.(s) 11-246393 filed: August 31, 1999
XX Preliminary Amendment
XX Notification of Change of Name and Address
XX A filing fee, calculated as shown below:

	(Col. 1)	(Col. 2)	Small Entity			Other Than A Small Entity	
FOR:	No. Filed	No. Extra	RATE	FEE		RATE	FEE
BASIC FEE				$345	or		$690
TOTAL CLAIMS	36 - 20 =	* 16	× 9 =		or	× 18 =	$288
INDEP CLAIMS	3 - 03 =	* 0	× 39 =		or	× 78 =	0
XX MULTIPLE DEPENDENT CLAIM PRESENTED			+130 =		or	+260 =	$260
			TOTAL		or		$1238

* If the difference in Col. 1 is less than zero, enter "0" in Col. 2

XX Check #299640 in the amount of $ 1278.00 to cover the filing fee and assignment recordation fee. In the event that the attached check is found to be insufficient, the Commissioner is hereby authorized to charge payment for any additional filing fees required under 37 CFR 1.16 associated with this communication or credit any over-payment to Deposit Account No. 01-2300.

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Respectfully submitted,

ARENT FOX KINTNER PLOTKIN & KAHN, PLLC

By: David T. Nikaido
Reg. No. 22,663

1050 Connecticut Avenue, N.W.
Suite 600
Washington, D. C. 20036-5339
Tel: (202) 857-6000
Fax: (202) 638-4810

DTN:mso

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re application of:

Shoutarou YODA

Serial No.: New Application

Filed: August 30, 2000

For: SPEECH RECOGNITION SYSTEM AND METHOD

PRELIMINARY AMENDMENT

Assistant Commissioner for Patents
Washington, D.C. 20231

August 30, 2000

Sir:

Prior to calculation of the filing fee and examination of this application, please amend the above-identified application as follows:

IN THE CLAIMS:

Please amend the claims as follows:

Claim 6, Line 15, delete "5", and insert in place therefor, -- 4 --.

Claim 12, Line 6, delete "11", and insert in place therefor, -- 10 --.

Claim 18, Line 26, delete "17", and insert in place therefor, -- 16 --.

REMARKS

The above amendment to the claims has been made to correct the multiple dependency of the claims and to put the application in better condition for examination.

In the event that any fees are due in connection with this paper, please charge our Deposit Account No. 01-2300.

Respectfully submitted,

ARENT FOX KINTNER PLOTKIN & KAHN, PLLC

David T. Nikaido
Reg. No. 22,663

Atty. Docket No.: P107156-00019

1050 Connecticut Avenue, N.W.
Suite 600
Washington, D. C. 20036-5339
Tel (202) 857-6000
Fax (202) 638-4810

DTN:mso

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TITLE OF THE INVENTION

SPEECH RECOGNITION SYSTEM AND METHOD

BACKGROUND OF THE INVENTION

FIELD OF THE INVENTION

The present invention relates to a speech recognition system capable of allowing electronic equipments to be controlled or manipulated with uttered voices or speeches, and a speech recognition method for use in such a speech recognition system.

DESCRIPTION OF THE RELATED ART

Known speech recognition systems of this type are adapted to electronic equipments, such as an on-board audio system and an on-board navigation system.

In an on-board audio system equipped with a speech recognition system, when a passenger says the name of a desired radio broadcasting station, for example, the speech recognition system recognizes the uttered speech and automatically tunes to the reception frequency of the radio broadcasting station based on the recognition result. This improves the operability of the on-board audio system and makes it easier for a passenger to use the on-board audio system.

This speech recognition system also has other capabilities that relieve a passenger of the burden of operating an MD (Mini Disc) player and/or CD (Compact Disc) player. When the passenger loads an information-carrying recording/reproducing medium, such as an MD disc, into the

MD player and says the title of a musical piece recorded on that recording/reproducing medium, for example, the speech recognition system recognizes the uttered speech and automatically plays the selected musical piece.

An on-board navigation system equipped with a speech recognition system is provided with a capability of recognizing a speech uttered by a driver or the like to specify the name of the destination and displaying a map showing the route from the present location to the destination. This capability allows the driver to concentrate on driving a vehicle, thus ensuring safer driving environments.

The above-described conventional speech recognition systems are designed to cope with a single person who utters words of instructions. The conventional speech recognition systems therefore have only a single microphone for inputting speeches provided at a location nearest to a driver who is very likely to use the microphone.

Other passengers who are seated far from the microphone should therefore utter large voices toward the microphone to secure a sufficient input voice level. To improve the speech recognition precision of such a speech recognition system, other passengers than the driver should also utter large voices toward the microphone to input uttered speeches into the microphone without being affected by noise in a vehicle.

SUMMARY OF THE INVENTION

Accordingly, it is an object of the present invention to provide a speech recognition system which has an improved operability and can allow more than one person to secure a sufficient input voice level without uttering large voices or without being affected by ambient noise.

It is another object of this invention to provide a speech recognition method for use in a speech recognition system, which improves the operability of the speech recognition system.

To achieve the first object, according to one aspect of this invention, there is provided a speech recognition system which comprises a plurality of voice pickup sections for picking up uttered voices; a determination section for determining a speech signal suitable for speech recognition from speech signals output from the plurality of voice pickup sections; and a speech recognizer for performing speech recognition based on the speech signal determined by the determination section.

According to another aspect of this invention, there is provided a speech recognition method for a speech recognition system having a plurality of voice pickup means for picking up voices, which comprises a determination step of determining a speech signal suitable for speech recognition from speech signals output from the plurality of voice pickup means; and a speech recognition step of performing speech recognition based on the speech signal determined by the determination step.

In the speech recognition system or speech recognition method, that of the speech signals output from the plurality of voice pickup sections (voice pickup means) whose speech level is equal to or higher than a predetermined speech level and continues over a predetermined period of time may be determined as the speech signal suitable for speech recognition.

It is preferable that the determination section (or step) acquires an average S/N value and average voice power of each of the speech signals output from the plurality of voice pickup sections (or voice pickup means) and determines that of the speech signal whose average S/N value and average voice power are greater than respective predetermined threshold values as the speech signal suitable for speech recognition.

In this case, it is preferable that the determination section determines a candidate order of those speech signals whose average S/N values and average voice powers are greater than the respective predetermined threshold values and which are candidates for the speech signal suitable for speech recognition, in accordance with the average S/N values and average voice powers; and the speech recognizer sequentially executes speech recognition on the candidates in accordance with the candidate order from a highest candidate to a lower one.

In any of the speech recognition system and method and their preferable modes, the determination section (or step)

treats those of the speech signals which are other than the speech signal suitable for speech recognition as noise signals.

In any of the speech recognition system and method and their preferable modes, of other speech signals than the speech signal suitable for speech recognition, that speech signal whose average S/N value and average voice power become minimum may be treated as a noise signal by the determination section.

With the above structures, when a speaker makes a desired speech, a speech signal and a noise signal suitable for speech recognition are automatically determined from the individual speech signals output from a plurality of voice pickup sections (or voice pickup means) and speech recognition is carried out based on the determined speech signal and noise signal. Accordingly, the speaker has only to utter words or voices without consciously making such a speech to a specific voice pickup section. This leads to an improved operability of the speech recognition system.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 is a block diagram illustrating the structure of a speech recognition system according to one embodiment of the present invention;

FIG. 2A is a plan view exemplifying the layout of microphones in an ordinary 4-seat vehicle;

FIG. 2B is a plan view showing another layout of microphones in an ordinary 4-seat vehicle;

FIG. 3A is a plan view exemplifying the layout of microphones in a wagon or the like;

FIG. 3B is a plan view showing another layout of microphones in a wagon or the like;

FIG. 4 is a block diagram showing the structures of a multiplexer, a demultiplexer and a storage section;

FIG. 5 is a timing chart for explaining the timings of sampling an input signal and storing sampled signals into a storage section;

FIGS. 6A through 6D are explanatory diagrams for explaining how to compute an average voice power, an average noise power and an average S/N value;

FIG. 7 is an explanatory diagram showing the structure of a speech condition table;

FIG. 8 is an explanatory diagram showing the structure of a noise selection table;

FIG. 9 is a flowchart for explaining the operation of the speech recognition system according to this embodiment;

FIG. 10 is a flowchart for further explaining the operation of the speech recognition system according to this embodiment; and

FIG. 11 is a block diagram illustrating the structure of a modification of the speech recognition system according to this embodiment.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

With reference to the accompanying drawings, a description will now be given of a preferred embodiment of

the present invention as adapted to a speech recognition system which can ensure voice- or speech-based control or manipulation of an electronic equipment installed in a vehicle, such as an on-board audio system or an on-board navigation system.

FIG. 1 is a block diagram illustrating the structure of a speech recognition system according to this embodiment of this invention. Referring to this diagram, the speech recognition system comprises a plurality of microphones M_1 to M_N as voice pickup means, a plurality of pre-circuits CC_1 to CC_N, a multiplexer 1, an A/D (Analog-to-Digital) converter (ADC) 2, a demultiplexer 3, a storage section 4, a speech detector 5, a data analyzer 6, a speech recognizer 7, a controller 8 and a speech switch 9.

The pre-circuits CC_1-CC_N, the multiplexer 1, the A/D converter 2, the demultiplexer 3, the storage section 4, the speech detector 5, the data analyzer 6 and the controller 8 constitute determination means which determines a speech signal and noise signal suitable for speech recognition.

The single speech switch 9 is provided in the vicinity of a driver seat, for example, on a front dash board or one end of a front door by the driver seat.

The controller 8 has a microprocessor (MPU), which controls the general operation of this speech recognition system. When the speech switch 9 is switched on, sending an ON signal SW to the microprocessor, the microprocessor causes the microphones M_1-M_N to initiate a voice pickup

operation.

The speech detector 5 has number-of-speeches counters FC_1-FC_N that are used to determine to which microphone an uttered speech is directed, though their details will be given in a later description of the operation of the speech recognition system.

The individual microphones M_1-M_N are provided at locations where it is easy to pick up speeches uttered by individual passengers, e.g., in the vicinity of the individual passenger seats including the driver seat.

In one example where four microphones M_1-M_4 are placed in a 4-seat vehicle, the microphones M_1 and M_2 are placed in front of the driver seat and the front passenger seat and the microphones M_3 and M_4 are placed in front of the rear passenger seats, e.g., the corresponding roof portions or at the back of the driver seat and the front passenger seat as shown in a plan view of FIG. 2A. This way, the individual microphones M_1-M_4 are associated with the respective passengers.

In another example as shown in a plan view of FIG. 2B, the microphones M_1 and M_2 may be placed in the front door by the driver seat and the front door by the front passenger seat and the microphones M_3 and M_4 are placed in the rear doors by the respective rear passenger seats, so that the individual microphones M_1-M_4 are associated with the respective passengers.

In a further example, the microphones M_1-M_4 may be

provided at combined locations shown in FIGS. 2A and 2B. Specifically, the microphone M_1 is placed in front of the driver seat as shown in FIG. 2A or in the front door by the driver seat as shown in FIG. 2B, so that a single microphone is provided for the driver who sits on the driver seat. Likewise, either the location shown in FIG. 2A or the location shown in FIG. 2B is selected for any of the remaining microphones M_2-M_4.

In the case of a wagon type vehicle or the like which holds a greater number of seats, for example, a greater number of microphones M_1-M_6 are provided in accordance with the seats and at the locations where it is easy to pick up speeches uttered by individual passengers, as shown in plan views of FIGS. 3A and 3B. Note that the microphones M_1-M_6 may be provided at combined locations shown in FIGS. 3A and 3B as per the aforementioned case of the 4-seat vehicle.

It is to be noted that the aforementioned microphone layouts have been given simply as examples, and are to be considered as illustrative and not restrictive. Actually, system information that is used in the speech recognition system of this invention is constructed beforehand in consideration of the characteristics of voice transmission from individual passengers to the respective microphones. Strictly speaking, therefore, the conditions for setting the microphones are not restricted at all. Further, the number of microphones can be determined to be equal to or smaller than the number of maximum passengers predetermined in

accordance with the type of a vehicle.

The layout of the individual microphones is not limited to a simple layout that makes the distances between the microphones to the respective passengers equal to one another. Those distances and the locations of the individual microphones may be determined based on the results of analysis of the voice characteristics in a vehicle previously acquired through experiments or the like in such a way that the characteristics of voice transmission from the microphones to the respective passengers become substantially the same.

Returning to FIG. 1, the microphones M_1-M_N are connected to the respective pre-circuits CC_1-CC_N, thus constituting N channels of signal processing systems.

Each of the pre-circuits CC_1-CC_N has an amplifier (not shown) which amplifies the amplitude level of the associated one of input speech signals S_1 to S_N, supplied from the microphones M_1-M_N, to the level that is suitable for signal processing, and a band-pass filter (not shown) which passes only a predetermined frequency component of the amplified input speech signal. The pre-circuits CC_1-CC_N supply input speech signals S_1' to S_N', which have passed the respective band-pass filters, to the multiplexer 1.

Each band-pass filter is set with a low cut-off frequency f_L (e.g., f_L = 100 Hz) for eliminating low-frequency noise included in the associated one of the input speech signals S_1-S_N and a high cut-off frequency f_H in

consideration of the Nyquist frequency. The low cut-off frequency f_L and high cut-off frequency f_H are set so that the frequency range of voices that human beings utter is included in the range between those two frequencies.

As shown in FIG. 4, the multiplexer 1 comprises analog switches AS_1 to AS_N for N channels. The input speech signals S_1'-S_N' from the pre-circuits CC_1-CC_N are supplied to the input terminals of the respective analog switches AS_1-AS_N whose output terminals are connected together to the A/D converter 2. In accordance with channel switch signals CH_1 to CH_N supplied from the controller 8, the analog switches AS_1-AS_N exclusively switch the input speech signals S_1'-S_N' and supply the switched input speech signals S_1'-S_N' to the A/D converter 2.

The A/D converter 2 convert the input speech signals S_1'-S_N', sequentially supplied from the multiplexer 1, to digital input data D_1 to D_N in synchronism with a predetermined sampling frequency f, and supplies the digital input data D_1-D_N to the demultiplexer 3.

The sampling frequency f is set by a sampling clock CK_{ADC} from the controller 8 and is determined in consideration of anti-aliasing. More specifically, the sampling frequency f is determined to be equal to or higher than approximately twice the high cut-off frequency f_H of the band-pass filter, and is set, for example, in a range of 8 kHz to 11 kHz.

The demultiplexer 3 comprises analog switches AW_1 to AW_N

for N channels, as shown in FIG. 4. The analog switches AW_1-AW_N have their input terminals connected together to the output terminal of the A/D converter 2 and their output terminals respectively connected to memory areas ME_1 to ME_N for N channels provided in the storage section 4. In accordance with the channel switch signals CH_1-CH_N supplied from the controller 8, the analog switches AW_1-AW_N exclusively switch the input data D_1-D_N and supply the switched input data D_1-D_N to the respective memory areas ME_1-ME_N.

Referring now to the timing chart in FIG. 5, the operations of the multiplexer 1, the A/D converter 2 and the demultiplexer 3 will be explained. When the speech switch 9 is set on, the resultant ON signal SW is received by the controller 8 which in turn outputs the sampling clock CK_{ADC} and the channel switch signals CH_1-CH_N.

The sampling clock CK_{ADC} has a pulse waveform which repeats the logical inversion N times during a period (sampling period) T which is the reciprocal, 1/f, of the sampling frequency f. The channel switch signals CH_1-CH_N have pulse waveforms which sequentially become logic "1" every period T/N of the sampling clock CK_{ADC}.

The multiplexer 1 exclusively performs switching between enabling and disabling of the input speech signals S_1'-S_N' in synchronism with the period T/N in which the channel switch signals CH_1-CH_N sequentially become logic "1". As a result, the input speech signals S_1'-S_N' are

sequentially supplied to the A/D converter 2 in synchronism with the period T/N to be converted to the digital data D_1-D_N. The demultiplexer 3 likewise exclusively performs switching between enabling and disabling of the input data D_1-D_N in synchronism with the period T/N in which the channel switch signals CH_1-CH_N sequentially become logic "1". Accordingly, the input data D_1-D_N from the A/D converter 2 are distributed and stored in the respective memory areas ME_1-ME_N in synchronism with the period T/N.

As sampling N channels of input speech signals S_1'-S_N' in the sampling period T (= 1/f) is repeated this way, it is possible to generate N channels of input data D_1-D_N with even the single A/D converter 2 in synchronism with the sampling frequency f and to store the input data D_1-D_N into the predetermined memory areas ME_1-ME_N, respectively.

The storage section 4, which is constituted by a semiconductor memory, has the aforementioned memory areas ME_1-ME_N for N channels. That is, the memory areas ME_1-ME_N are provided in association with the microphones M_1-M_N.

As shown in FIG. 4, each of the memory areas ME_1-ME_N has a plurality of frame areas MF_1, MF_2 and so forth for storing the associated one of the input data D_1-D_N frame by frame of a predetermined number of samples.

Referring to the memory area ME_1, for example, the frame areas MF_1, MF_2 and so forth sequentially store the input data D_1 supplied from the demultiplexer 3 by a predetermined number of samples (256 samples in this

embodiment) in accordance with an address signal ADR_1 from the controller 8. That is, every 256 samples of the input data D_1 are stored in each frame area MF_1, MF_2 or the like in each frame period TF which is 256 x T as shown in FIG. 5. Input data for one frame period (1TF), which is stored in each frame area MF_1, MF_2 or the like, is called "frame data".

Likewise, the input data D_2-D_N are stored, 256 samples each, in the frame area MF_1, MF_2 and so forth in the remaining memory areas ME_2-ME_N in each frame period TF.

The speech detector 5 and the data analyzer 6 are constituted by a DSP (Digital Signal Processor).

Every time frame data is stored in the frame area MF_1, MF_2 and so forth in each of the memory areas ME_1-ME_N, the speech detector 5 computes the LPC (Linear Predictive Coding) residual of the latest frame data and determines if the computed value is equal to or greater than a predetermined threshold value THD1. When the computed value becomes equal to or greater than the predetermined threshold value THD1, the speech detector 5 determines that the latest frame data is speech frame data produced from a speech. When the computed value is smaller than the predetermined threshold value THD1, the speech detector 5 determines that the latest frame data is input data that has not been produced from a speech, i.e., noise frame data that has been produced by noise in a vehicle.

When the computed LPC residual value becomes equal to or greater than the predetermined threshold value THD1 over

three frame periods (3TF), the speech detector 5 settles that the frame data over the three frame periods (3TF) is definitely speech frame data produced from a speech and transfers speech detection data DCT1 indicative of the result of the decision to the controller 8.

More specifically, the LPC residuals of frame data stored in the individual frame area MF_1, MF_2 and so forth in each of the memory areas ME_1-ME_N are individually computed channel by channel, and each channel-by-channel computed LPC residual value is compared with the threshold value THD1 to determine, channel by channel, if the frame data is speech frame data produced from a speech.

Given that ε_1 is the computed LPC residual value of the first channel associated with the microphone M_1, ε_2 is the computed LPC residual value of the second channel associated with the microphone M_2 and likewise ε_3 to ε_N are the computed LPC residual values of the third to N-th channels respectively associated with the microphones M_3-M_N, the computed values ε_1-ε_N are compared with the threshold value THD1. The frame data that corresponds to the channel whose computed LPC residual value becomes equal to or greater than the threshold value THD1 is determined as speech frame data that has been generated from a speech. Further, the speech frame data that corresponds to the channel whose computed LPC residual value becomes equal to or greater than the threshold value THD1 over three frame periods (3TF) is settled as speech frame data that is definitely generated

from a speech.

When a speech has been directed to the microphone M_1 and the uttered voices have not been input to the remaining microphones M_2-M_N, for example, only the frame data that is stored in the memory area ME_1 of the channel associated with the microphone M_1 is determined and settled as speech frame data that has been produced from the speech, and the frame data stored in the memory areas ME_2-ME_N associated with the remaining microphones M_2-M_N are determined as noise frame data generated from noise in the vehicle.

When a speech has been directed to the microphone M_1 and the uttered voices have reached the microphone M_2 but not the remaining microphones M_3-M_N, for example, the frame data stored in the memory areas ME_1 and ME_2 of the channels associated with the microphones M_1 and M_2 are both determined and settled as speech frame data produced from the speech, and the frame data stored in the memory areas ME_3-ME_N associated with the remaining microphones M_3-M_N are determined as noise frame data.

In the above-described manner, the speech detector 5 computes the LPC residual of each of the frame data stored in the memory areas ME_1-ME_N, compares it with the threshold value THD1 to determine if uttered voices have been input to any microphone and determine the frame period in which the uttered voices have been input, and transfers the speech detection data DCT1 having information on those decisions to the controller 8.

The speech detection data DCT1 is transferred to the controller 8 as predetermined code data which indicates the memory area where speech frame data has been stored over the aforementioned three frames or more (hereinafter this memory area will be called "speech memory channel") and its frame area (hereinafter called "speech memory frame").

Specifically, the speech detection data DCT1 has an ordinary data structure of, for example, DCT1{CH_1(TF_1, TF_2-TF_m), CH_2(TF_1, TF_2-TF_m), ..., CH_N(TF_1, TF_2-TF_m)}. CH_1-CH_N are flag data representing the individual channels, and TF_1, TF_2-TF_m are flag data corresponding to the individual frame areas MF_1, MF_2-MF_m.

When an uttered speech is input only to the microphone M_1 and speech frame data is stored in the third and subsequent frame areas MF_3, MF_4 and so forth, speech detection data DCT of binary codes of DCT1{1(0,0,1,1-1), 0(0,0,0-0), ..., 0(0,0,0-0) is transferred to the controller 8.

When the speech detection data DCT1 is transferred to the controller 8, the controller 8 generates control data CNT1 indicating the speech memory channel and speech memory frame based on the speech detection data DCT1, and sends the control data CNT1 to the data analyzer 6

The data analyzer 6 comprises an optimal-speech determining section 6a, a noise determining section 6b, an average-S/N computing section 6c, an average-voice-power computing section 6d, an average-noise-power computing

section 6e, a speech condition table 6f and a noise selection table 6g. When receiving the control data CNT1 from the controller 8, the data analyzer 6 initiates a process of determining speech frame data and noise frame data suitable for speech recognition.

The average-voice-power computing section 6d acquires information on the speech memory channel and speech memory frame from the control data CNT1, reads speech frame data from the memory area that corresponds to those speech memory channel and speech memory frame and computes average voice power P(n) of the speech frame data channel by channel. The variable n in the average voice power P(n) indicates a channel number.

When speech frame data is stored in the memory areas ME_1-ME_4 corresponding to the channels CH_1-CH_4 as shown in FIGS. 6A to 6D, for example, the average voice power P(1) to P(4) of plural pieces of speech frame data corresponding to a plurality of predetermined frame periods (m_2 x TF) from a time t_s at which a speech has started are computed channel by channel. The average voice power P(n) is computed by obtaining the sum of squares of speech frame data in the frame periods (m_2 x TF) and then dividing the sum by the number of the frame periods (m_2 x TF).

The average-noise-power computing section 6e acquires information on the speech memory channel and speech memory frame from the control data CNT1, reads noise frame data preceding the speech frame data by a plurality of frame

periods (m_1 x TF) from the memory area that corresponds to those speech memory channel and speech memory frame and computes average noise power NP(n) of the noise frame data channel by channel. The variable n in the average noise power NP(n) indicates a speech channel, and the average noise power NP(n) is computed by obtaining the sum of squares of noise frame data in the frame periods (m_1 x TF) and then dividing the sum by the number of the frame periods (m_1 x TF).

When speech frame data is stored in the memory areas ME_1-ME_4 corresponding to the channels CH_1-CH_4 as shown in FIGS. 6A to 6D, for example, the average noise power NP(n) of plural pieces of noise frame data preceding by a plurality of frame periods (m_1 x TF) from the time t_s at which a speech has started (at which storage of the speech frame data has started) are computed.

The average-S/N computing section 6c computes an average S/N value SN(n) which represents the value of the signal-to-noise ratio for each speech channel based on the average voice power P(n) computed by the average-voice-power computing section 6d and the average noise power NP(n) computed by the average-noise-power computing section 6e.

In the case where the channels CH_1-CH_4 are speech channels as shown in FIGS. 6A to 6D, for example, the average S/N values SN(1) to SN(4) of the individual channels CH_1-CH_4 are computed from the following equations 1 to 4.

$$SN(1) = P(1)/NP(1) \qquad \ldots (1)$$

$$SN(2) = P(2)/NP(2) \quad \ldots (2)$$

$$SN(3) = P(3)/NP(3) \quad \ldots (3)$$

$$SN(4) = P(4)/NP(4) \quad \ldots (4)$$

Logarithmic values of the average S/N values SN(1) to SN(4) computed from the equations 1 to 4 may be taken as the average S/N values SN(1)-SN(4) of the individual channels CH_1-CH_4.

The optimal-speech determining section 6a compares the average S/N value SN(n) acquired by the average-S/N computing section 6c with a predetermined threshold value THD2, and compares the average voice power P(n) acquired by the average-voice-power computing section 6d with a predetermined threshold value THD3. The optimal-speech determining section 6a then collates the results of the comparison with the speech condition table 6f shown in FIG. 7 to determine which channel of speech frame data is suitable for the speech recognition process.

As shown in FIG. 7, the speech condition table 6f is storing reference data for ranking speech frame data in accordance with the relationship between the average S/N value and the threshold value THD2 and the relationship between the average voice power and the threshold value THD3. Referring to the speech condition table 6f based on the comparison results, the optimal-speech determining section 6a ranks the speech frame data suitable for speech recognition and determines the speech frame data of the highest rank as the one suitable for speech recognition.

Specifically, the optimal-speech determining section 6a determines the speech frame data whose average S/N value is equal to or greater than the threshold value THD2 and whose average voice power is equal to or greater than the threshold value THD3 as a rank 1 (Rnk1), determines the speech frame data whose average S/N value is equal to or greater than the threshold value THD2 and whose average voice power is less than the threshold value THD3 as a rank 2 (Rnk2), determines the speech frame data whose average S/N value is smaller than the threshold value THD2 and whose average voice power is equal to or greater than the threshold value THD3 as a rank 3 (Rnk3), and determines the speech frame data whose average S/N value is smaller than the threshold value THD2 and whose average voice power is less than the threshold value THD3 as a rank 3 (Rnk3).

Further, the optimal-speech determining section 6a determines the speech frame data in all the channels of speech frame data whose average S/N value and average voice power become maximum as a rank 0 (Rnk0).

Then, the optimal-speech determining section 6a determines the speech frame data that becomes the rank 0 (Rnk0) as a candidate most suitable for speech recognition (first candidate). Further, the optimal-speech determining section 6a determines the speech frame data that becomes the rank 1 (Rnk1) as the next candidate suitable for speech recognition (second candidate). When there are a plurality of channels whose speech frame data become the rank 1 (Rnk1),

those speech frame data which have greater average S/N values and greater average voice powers are determined as candidates of higher ranks.

Further, the optimal-speech determining section 6a removes the speech frame data that correspond to the rank 2 (Rnk2) to the rank 4 (Rnk4) from the targets for speech recognition, considering that they are unsuitable for speech recognition.

In short, the optimal-speech determining section 6a compares the average S/N value SN(n) and the average voice power P(n) with the threshold values THD2 and THD3 respectively, collates the comparison results with the speech condition table 6f shown in FIG. 7 to determine the speech frame data that is suitable for speech recognition, and then puts a priority order or ranking to speech frame data suitable for speech recognition. Then, the optimal-speech determining section 6a transfers speech candidate data DCT2 indicating the ranking to the controller 8.

The noise determining section 6b collates combinations of all the ranks for N channels that are acquired by the optimal-speech determining section 6a with the noise selection table 6g shown in FIG. 8, and determines any channel for which the ranking combination has a match as a noise channel.

When the ranks of the individual channels starting at the first channel CH_1 are (Rnk0), (Rnk1), (Rnk2), (Rnk1), ..., for example, the noise determining section 6b determines the

third channel CH_3 as a noise channel. Then, the noise determining section 6b sends noise candidate data DCT3 to the controller 8.

When the optimal-speech determining section 6a determines a candidate of speech frame data suitable for speech recognition, the noise determining section 6b determines a noise channel corresponding to the candidate of speech frame data suitable for speech recognition by referring to the individual "cases" in FIG. 8. Accordingly, a candidate of speech frame data suitable for speech recognition and noise data obtained by the microphone that has picked up noise are determined in association with each other.

The individual cases 1, 2, 3 and so forth in the noise selection table 6g in FIG. 8 are preset based on the results of experiments on the voice characteristics obtained when passengers actually uttered voices at various positions in a vehicle in which all the microphones M_1-M_N were actually installed.

When the speech candidate data DCT2 and the noise candidate data DCT3 are supplied to the controller 8, the controller 8 accesses that of the memory areas ME_1-ME_N which corresponds to the channel of the first candidate based on the speech candidate data DCT2, reads the speech frame data most suitable for speech recognition and supplies it to the speech recognizer 7.

The speech recognizer 7 performs known processes, such

as SS (Spectrum Subtraction), echo canceling, noise canceling and CMN, based on the speech frame data and noise frame data supplied from the storage section 4 to thereby eliminate a noise component from the speech frame data, performs speech recognition based on the noise-component removed speech frame data and outputs data Dout representing the result of speech recognition.

If an adequate speech recognition result is not acquired from the speech recognition performed by the speech recognizer 7 based on speech frame data and the noise frame data suitable for speech recognition, the controller 8 accesses the memory area that corresponds to the channel of the next candidate suitable for speech recognition and transfers the corresponding speech frame data to the speech recognizer 7. Thereafter, the controller 8 supplies speech frame data of the channels of subsequent candidates in order to the speech recognizer 7 until the adequate speech recognition result is acquired.

An example of the operation of this speech recognition system which has the above-described structure will be discussed with reference to the flowcharts shown in FIGS. 9 and 10. FIG. 9 illustrates an operational sequence from the pickup of sounds with the microphones M_1-M_N to the storage of the input data D_1-D_N into the storage section 4 as frame data, and FIG. 10 illustrates the operation at the time the data analyzer 6 determines optimal speech frame data and noise frame data.

In FIG. 9, the speech recognition system stands by until the speech switch 9 is switched on in step 100. Upon occurrence of the ON event of the speech switch 9, the flow goes to step 102 to perform initialization. This initializing process clears a count value n of a channel-number counter, a count value m of a frame-number counter and all values F(1) to F(N) of the number-of-speeches counters FC_1-FC_N, all provided in the controller 8.

The channel-number counter is provided to designate each of the channels of the microphones M_1-M_N with the count value n. The frame-number counter is provided to designate the number (address) of each of the frame areas MF_1, MF_2, MF_3 and so forth, provided in the each of the memory areas ME_1-ME_N, with the count value m.

N number-of-speeches counters FC_1-FC_N are provided in association with the individual channels. That is, the first number-of-speeches counter FC_1 is provided in association with the first channel, the second number-of-speeches counter FC_2 is provided in association with the second channel, and so forth to the N-th number-of-speeches counter FC_N provided in association with the N-th channel. The number-of-speeches counters FC_1-FC_N are used to determine whether or not an LPC residual ε_n greater than the threshold value THD1 has consecutively continued over three or more frames and to determine the channel for which the LPC residual ε_n has continued over three or more frames. The number-of-speeches counters FC_1-FC_N are also used to

determine, as a speech-input channel, the channel for which the LPC residual ε_n has continued over three or more frames.

In the next step 104, the first frame area MF_1 of each of the memory areas ME_1-ME_N is set. That is, the number, m, of the frame area is set to m = 1.

In subsequent steps 106 and 108, the microphones M_1-M_N start picking up sounds and the input data D_1-D_N acquired by the voice pickup are stored in the individual first frame areas MF_1 of the memory areas ME_1-ME_N frame by frame.

When one frame of input data D_1-D_N is stored, the memory area ME_1 that corresponds to the first (n = 1) channel is designated in step 110, and the LPC residual ε_n (n = 1) of frame data stored in the first (m = 1) frame area MF_1 of the memory area ME_1 is computed in step 112.

In the next step 114, the LPC residual ε_n is compared with the threshold value THD1. When $\varepsilon_n \geqq$ THD1, the flow goes to step 116 to increment (or adds "1" to) the value F(1) of the number-of-speeches counter FC_1 corresponding to the first channel by "1". When $\varepsilon_n <$ THD1, the flow goes to step 118 to clear the value F(1) of the number-of-speeches counter FC_1.

When ε_n becomes equal to or greater than THD1 ($\varepsilon_n \geqq$ THD1), therefore, the value F(1) of the number-of-speeches counter FC_1 becomes "1" which indicates that one frame of speeches has been input to the microphone M_1 of the first channel.

When ε_n becomes smaller than THD1 ($\varepsilon_n <$ THD1), on the

other hand, the value F(1) of the number-of-speeches counter FC_1 is cleared to "0" which indicates that no speeches have been input to the microphone M_1 of the first channel.

Next, it is checked if n is equal to N (n = N) in step 120 to determine whether the LPC residual ε_n in every channel has been computed. When n = N is not met, the flow goes to step 122 to make n = n + 1 to set the next channel, and the sequence of processes from step 112 is repeated. That is, by repeating the processes of steps 112 to 122, the LPC residual ε_n of frame data stored in the frame area MF_1 of each of the memory areas ME_1-ME_N is compared with the threshold value THD1. When the LPC residual ε_n becomes equal to or greater than the threshold value THD1, the value F(n) of the number-of-speeches counter FC_1 corresponding to that channel number n is incremented by "1".

When n = N is met in the aforementioned step 120, it is determined that the processing for all the channels has been completed, then the flow proceeds to step 124.

In step 124, it is determined if any one of the values F(1) to F(N) of the number-of-speeches counters FC_1-FC_N has become equal to or greater than "3". If there is no such a count value, i.e., if any of the values F(1) to F(N) is equal to or smaller than "2", the flow goes to step 126.

In step 126, the individual second frame areas MF_2 of the memory areas ME_1-ME_N 1 are set by setting m = m + 1. Then, the processes of steps 106 to 124 are repeated.

Accordingly, the input data is stored in each frame

area MF_2 (steps 106 and 108), the LPC residual ε_n of each frame data stored in each frame area MF_2 is compared with the threshold value THD1 (steps 110 to 114), and each of the values F(1) to F(N) of the number-of-speeches counters FC_1-FC_N is incremented or cleared based on the comparison results.

In step 124, it is determined again if any one of the values F(1) to F(N) of the number-of-speeches counters FC_1-FC_N has become equal to or greater than "3". If there is no such a count value, the flow goes to step 126 to set m = m + 1 so that the next frame areas MF_3 of the memory areas ME_1-ME_N 1 are set. Then, the processes of steps 106 to 124 are repeated.

As the processes of steps 106 to 124 are repeated and at least one of the values F(1) to F(N) of the number-of-speeches counters FC_1-FC_N becomes equal to or greater than "3", the flow proceeds to step 128.

In other words, in step 124, the values F(1) to F(N) of the number-of-speeches counters FC_1-FC_N are checked and only when the LPC residual ε_n greater than the threshold value THD1 consecutively continues over three or more frames, frame data stored in the memory area corresponding to that channel is determined and settled as speech frame data.

In the next step 128, it is determined if the value of the number-of-speeches counter for which it was determined the LPC residual ε_n greater than the threshold value THD1 consecutively continued over three or more frames has reached "5". If that value has not reached "5" yet, the

process in step 126 is carried out after which the processes of steps 106 to 128 are repeated.

There may be a case where when the value F(n) of the number-of-speeches counter that corresponds to a given channel n becomes "3", the value of the number-of-speeches counters corresponding to the remaining channels is "1" or "2". In this case, frame data stored in the memory areas corresponding to the remaining channels are likely to be also speech frame data.

To cope with this case, therefore, the processes of steps 106 to 128 are repeated twice to check if the frame data stored in the memory areas corresponding to the remaining channels are speech frame data.

When the decision in step 128 is "YES", the flow goes to step 130 where the speech detection data DCT1 which has information on the memory area where speech frame data is stored and the memory area where noise frame data is stored is transferred to the controller 8. The flow then proceeds to a routine illustrated in FIG. 10.

When the operation goes to the routine illustrated in FIG. 10, the average voice power P(n), the average noise power NP(n) and the average S/N value SN(n) for each channel are computed first in step 200. Next, a candidate of speech frame data suitable for speech recognition is determined based on the speech condition table 6f shown in FIG. 7 in step 202. In the next step 204, noise frame data suitable for speech recognition is determined based on the noise

selection table 6g shown in FIG. 8.

In step 206, the speech candidate data DCT2 that indicates the candidate of speech frame data suitable for speech recognition and the noise candidate data DCT3 that indicates the noise frame data are sent to the controller 8 from the data analyzer 6. In other words, the speech candidate data DCT2 and the noise candidate data DCT3 inform the controller 8 of the candidate of speech frame data suitable for speech recognition and noise frame data suitable for speech recognition associated with that candidate.

In the next step 208, the speech recognizer 7 read the speech frame data and noise frame data most suitable for speech recognition from the storage section 4, performs speech recognition on the read speech frame data and noise frame data, and terminates a sequence of speech recognition processes when an adequate speech recognition result is acquired.

When no adequate speech recognition result is acquired, on the other hand, the speech recognizer 7 checks in step 212 if there are next candidates of speech frame data and noise frame data, reads the next candidates of speech frame data and noise frame data, if present, from the storage section 4 and repeats the sequence of processes starting at step 208. When no adequate speech recognition result is obtained even after re-execution of the speech recognition, the speech recognizer 7 likewise reads next candidates of

speech frame data and noise frame data from the storage section 4 and repeats the sequence of processes in steps 208 to 212 until the adequate speech recognition result is obtained.

According to this embodiment, as apparent from the above, a plurality of microphones M_1-M_N for inputting voices are placed in a vehicle and speech frame data and noise frame data suitable for speech recognition are automatically extracted from those speech frame data and noise frame data that are picked up by the microphones M_1-M_N and are subjected to speech recognition. This speech recognition system can therefore provide a plurality of speakers (passengers) with a better operability than the conventional speech recognition system that is designed for a single speaker.

When one of a plurality of passengers directs a desired speech to a certain microphone (e.g., M_1), the uttered speech may generally be picked up by the other microphones (M_2-M_N) so that it is difficult to determine which microphone has actually been intended to pick up the uttered speech. According to this embodiment, however, speech frame data and noise frame data suitable for speech recognition are automatically extracted by using the speech condition table 6f and the noise selection table 6g, respectively shown in FIGS. 7 and 8, and speech recognition is carried out based on the extracted speech frame data and noise frame data. This makes it possible to associate the passenger who has made the speech with the microphone (e.g., M_1) close to that

passenger with a very high probability.

Accordingly, this speech recognition system automatically specifies a passenger who tries to perform a voice-based manipulation of an electronic equipment installed in a vehicle and allows the optimal microphone (close to the passenger) to pick up the uttered speech. This can improve the speech recognition precision. With the use of this speech recognition system, a passenger requires a special manipulation but merely needs to utter words to give this or her voiced instruction through the appropriate microphone, so that this speech recognition system is considerably easy to use.

Suppose that while one or more passengers who do not intend to perform a voice-based manipulation of an on-board electronic equipment are making a conversation or the like, one person utters words to perform such a voice-based manipulation. Even in this case, the conversation or the like made by the passengers who are not performing the voice-based manipulation is determined as noise and eliminated from consideration by automatically extracting speech frame data and noise frame data suitable for speech recognition by using the speech condition table 6f and the noise selection table 6g, respectively shown in FIGS. 7 and 8, and then carrying out speech recognition based on the extracted speech frame data and noise frame data. This can provide a speech recognition system which is not affected by a conversation or the like taking place in a vehicle around

and which is very easy to use.

Although this embodiment is provided with the single speech switch 9, shown in FIG. 1, which is switched on by, for example, a driver, this invention is not limited to this particular structure. For example, a plurality of microphones M_1-M_N may be respectively provided with speech switches TK_1 to TK_N as shown in the block diagram in FIG. 11, so that when one of the speech switches is set on, the controller 8 allows the microphone that corresponds to the activated speech switch to pick up words and determines that the remaining microphones corresponding to the inactive speech switches have picked up noise in the vehicle.

This modified structure can specify the microphone that has picked up an uttered speech and the microphones that have picked up noise before speech recognition. This can shorten the processing time for easily determining speech data and noise data most suitable for speech recognition.

Further, the structure shown in FIG. 1 and the structure shown in FIG. 11 may be combined as needed. Specifically, speech switches smaller in number than the microphones M_1-M_N may be placed at adequate locations in a vehicle so that when one of the speech switches is set on, the controller 8 detects the event and initiates speech recognition. In this case, the speech switches do not completely correspond one-to-one to the microphones M_1-M_N, so that while speech recognition is carried out with the structure shown in FIG. 1, the microphone that has picked up

an uttered speech and the microphones that have picked up noise before speech recognition can be specified before speech recognition. This can shorten the processing time for determining speech data and noise data suitable for speech recognition.

In the case where the structure in FIG. 1 is adapted to the case where speech switches smaller in number than the microphones M_1-M_N are provided, each speech switch may be determined as the layout range for the associated microphone or microphones and one or more microphones belonging to each layout range may be specified previously depending on which speech switch has been set on. With this structure, those which are suitable for speech recognition have only to be extracted from pre-specified single or plural speech frame data and noise frame data, thus making it possible to shorten the processing time.

Although the foregoing description of this embodiment and modifications has been given of a speech recognition system adapted to an on-board electronic equipment, the speech recognition system of this invention can also be adapted to other types of electronic apparatuses, such as a general-purpose microcomputer system and a so-called word processor, to enable voice-based entry of sentences or voice-based document edition.

According to this invention, in short, when a speaker makes a desired speech, a speech signal and a noise signal suitable for speech recognition are automatically determined

from the individual speech signals output from a plurality of voice pickup sections (or voice pickup means) and speech recognition is carried out based on the determined speech signal and noise signal. Accordingly, the speaker has only to utter words or voices without consciously making such a speech to a specific voice pickup section. This leads to an improved operability of the speech recognition system.

Although only one embodiment of the present invention and some modifications thereof have been described herein, it should be apparent to those skilled in the art that the present invention may be embodied in many other specific forms without departing from the spirit or scope of the invention. Therefore, the present examples and embodiment are to be considered as illustrative and not restrictive and the invention is not to be limited to the details given herein, but may be modified within the scope of the appended claims.

What is claimed is:

1. A speech recognition system comprising:

a plurality of voice pickup means for picking up uttered voices;

determination means for determining a speech signal suitable for speech recognition from speech signals output from said plurality of voice pickup means; and

speech recognition means for performing speech recognition based on said speech signal determined by said determination means.

2. The speech recognition system according to claim 1, wherein that of said speech signals output from said plurality of voice pickup means whose speech level is equal to or higher than a predetermined speech level and continues over a predetermined period of time is determined as said speech signal suitable for speech recognition.

3. The speech recognition system according to claim 1, wherein said determination means acquires an average S/N value and average voice power of each of said speech signals output from said plurality of voice pickup means and determines that of said speech signal whose average S/N value and average voice power are greater than respective predetermined threshold values as said speech signal suitable for speech recognition.

4. The speech recognition system according to claim 3, wherein said determination means determines a candidate order of those speech signals whose average S/N values and

average voice powers are greater than said respective predetermined threshold values and which are candidates for said speech signal suitable for speech recognition, in accordance with said average S/N values and average voice powers; and

said speech recognition means sequentially executes speech recognition on said candidates in accordance with said candidate order from a highest candidate to a lower one.

5. The speech recognition system according to any one of claims 1 to 4, wherein said determination means treats those of said speech signals which are other than said speech signal suitable for speech recognition as noise signals.

6. The speech recognition system according to any one of claims 1 to 5, wherein of other speech signals than said speech signal suitable for speech recognition, that speech signal whose average S/N value and average voice power become minimum is treated as a noise signal by said determination means.

7. A speech recognition system comprising:

a plurality of voice pickup sections for picking up uttered voices;

a determination section for determining a speech signal suitable for speech recognition from speech signals output from said plurality of voice pickup sections; and

a speech recognizer for performing speech recognition based on said speech signal determined by said determination

section.

8. The speech recognition system according to claim 7, wherein that of said speech signals output from said plurality of voice pickup sections whose speech level is equal to or higher than a predetermined speech level and continues over a predetermined period of time is determined as said speech signal suitable for speech recognition.

9. The speech recognition system according to claim 7, wherein said determination section acquires an average S/N value and average voice power of each of said speech signals output from said plurality of voice pickup sections and determines that of said speech signal whose average S/N value and average voice power are greater than respective predetermined threshold values as said speech signal suitable for speech recognition.

10. The speech recognition system according to claim 9, wherein said determination section determines a candidate order of those speech signals whose average S/N values and average voice powers are greater than said respective predetermined threshold values and which are candidates for said speech signal suitable for speech recognition, in accordance with said average S/N values and average voice powers; and

said speech recognizer sequentially executes speech recognition on said candidates in accordance with said candidate order from a highest candidate to a lower one.

11. The speech recognition system according to any one

of claims 7 to 10, wherein said determination section treats those of said speech signals which are other than said speech signal suitable for speech recognition as noise signals.

12. The speech recognition system according to any one of claims 7 to 11, wherein of other speech signals than said speech signal suitable for speech recognition, that speech signal whose average S/N value and average voice power become minimum is treated as a noise signal by said determination section.

13. A speech recognition method for a speech recognition system having a plurality of voice pickup means for picking up voices, comprising:

a voice pickup step of picking up uttered voices using said plurality of voice pickup means;

a determination step of determining a speech signal suitable for speech recognition from speech signals output from said plurality of voice pickup means; and

a speech recognition step of performing speech recognition based on said speech signal determined by said determination step.

14. The speech recognition method according to claim 13, wherein that of said speech signals output from said plurality of voice pickup means whose speech level is equal to or higher than a predetermined speech level and continues over a predetermined period of time is determined as said speech signal suitable for speech recognition.

15. The speech recognition method according to claim 13, wherein said determination step includes a step of acquiring an average S/N value and average voice power of each of said speech signals output from said plurality of voice pickup means and determining that of said speech signal whose average S/N value and average voice power are greater than respective predetermined threshold values as said speech signal suitable for speech recognition.

16. The speech recognition method according to claim 15, wherein said determination step further includes a step of determining a candidate order of those speech signals whose average S/N values and average voice powers are greater than said respective predetermined threshold values and which are candidates for said speech signal suitable for speech recognition, in accordance with said average S/N values and average voice powers; and

said speech recognition step sequentially executes speech recognition on said candidates in accordance with said candidate order from a highest candidate to a lower one.

17. The speech recognition method according to any one of claims 13 to 16, wherein said determination step includes a step of treating those of said speech signals which are other than said speech signal suitable for speech recognition as noise signals.

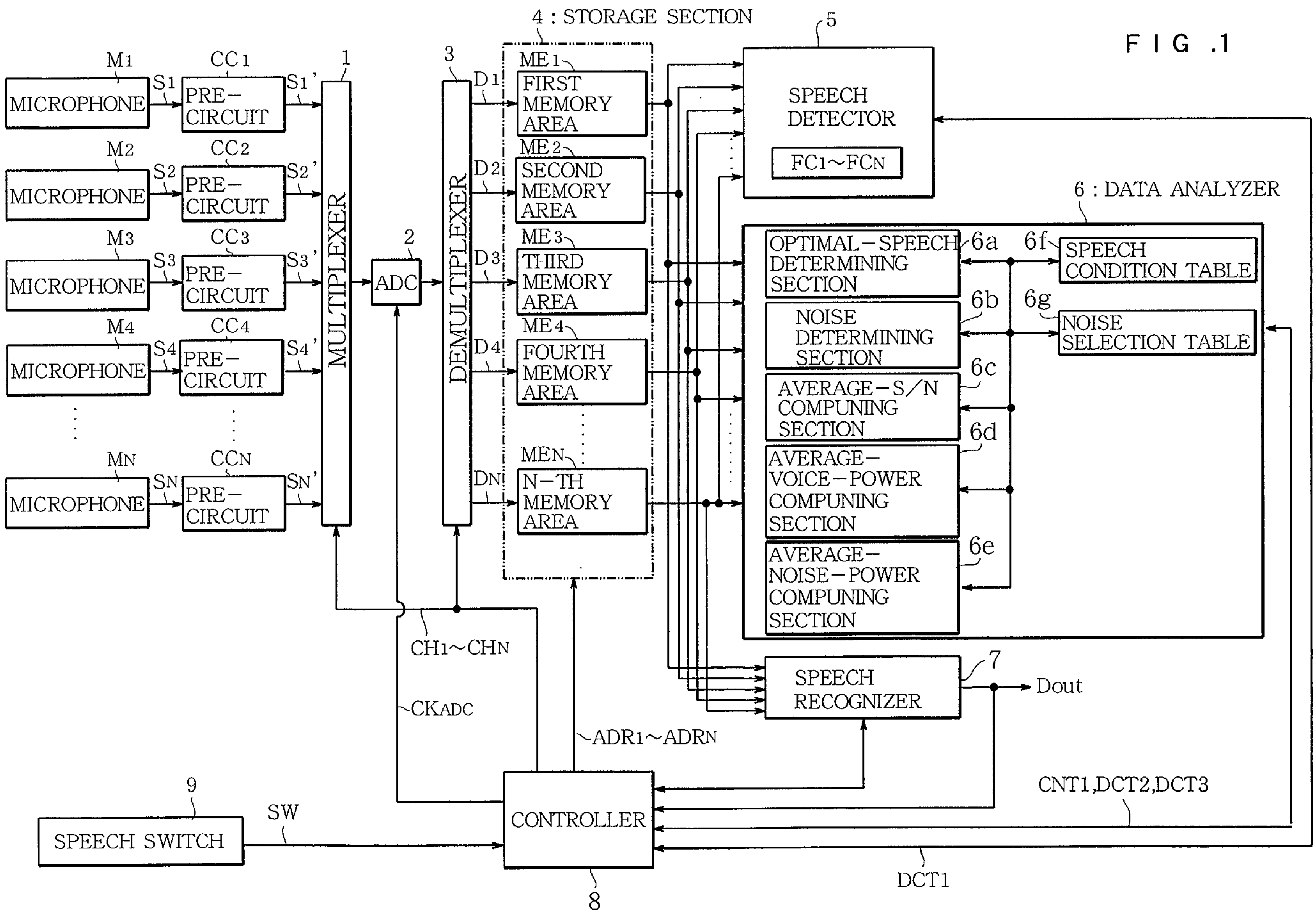
18. The speech recognition method according to any one of claims 13 to 17, wherein of other speech signals than said speech signal suitable for speech recognition, that

speech signal whose average S/N value and average voice power become minimum is treated as a noise signal in said determination step.

ABSTRACT OF THE DISCLOSURE

Disclosed are a speech recognition system which comprises the following components, and a speech recognition method for this speech recognition system. The speech recognition system comprises a plurality of voice pickup sections for picking up uttered voices, a determination section for determining a speech signal suitable for speech recognition from speech signals output from the plurality of voice pickup sections, and a speech recognizer for performing speech recognition based on the speech signal determined by the determination section.

F I G .1



4 : STORAGE SECTION
5
MICROPHONE
M1
M2
M3
M4
MN
S1
S2
S3
S4
SN
CC1
CC2
CC3
CC4
CCN
PRE-CIRCUIT
S1'
S2'
S3'
S4'
SN'
1
MULTIPLEXER
2
ADC
3
DEMULTIPLEXER
D1
D2
D3
D4
DN
ME1
FIRST MEMORY AREA
ME2
SECOND MEMORY AREA
ME3
THIRD MEMORY AREA
ME4
FOURTH MEMORY AREA
MEN
N-TH MEMORY AREA
SPEECH DETECTOR
FC1~FCN
6 : DATA ANALYZER
OPTIMAL-SPEECH DETERMINING SECTION
6a
6f
SPEECH CONDITION TABLE
NOISE DETERMINING SECTION
6b
6g
NOISE SELECTION TABLE
AVERAGE-S/N COMPUNING SECTION
6c
AVERAGE-VOICE-POWER COMPUNING SECTION
6d
AVERAGE-NOISE-POWER COMPUNING SECTION
6e
SPEECH RECOGNIZER
7
Dout
CH1~CHN
CKADC
ADR1~ADRN
CNT1,DCT2,DCT3
9
SPEECH SWITCH
SW
CONTROLLER
8
DCT1

FIG. 2 A

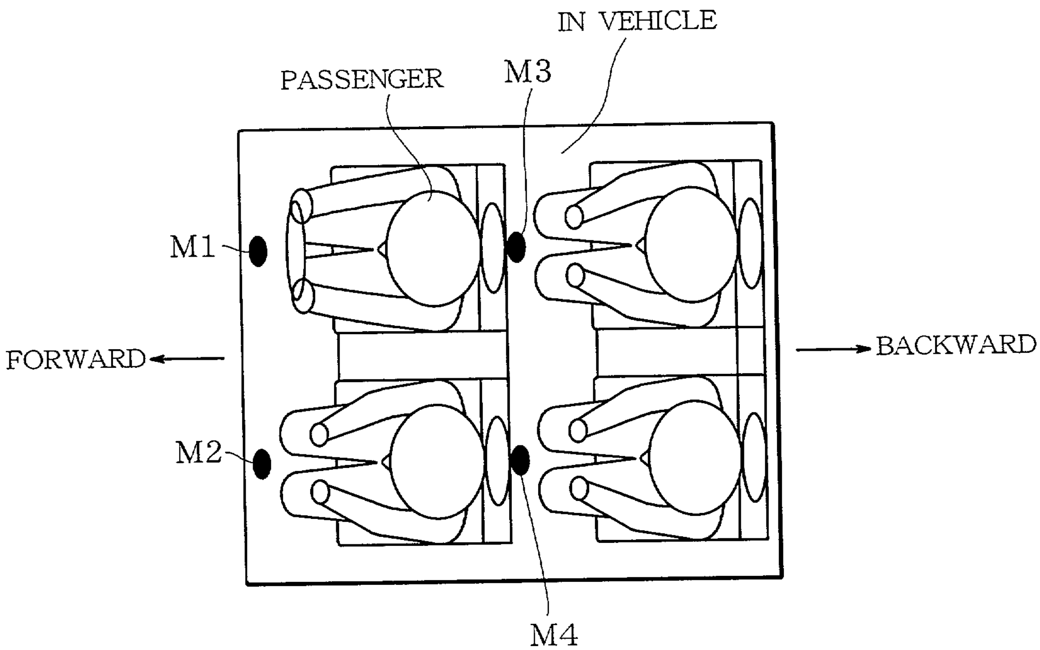
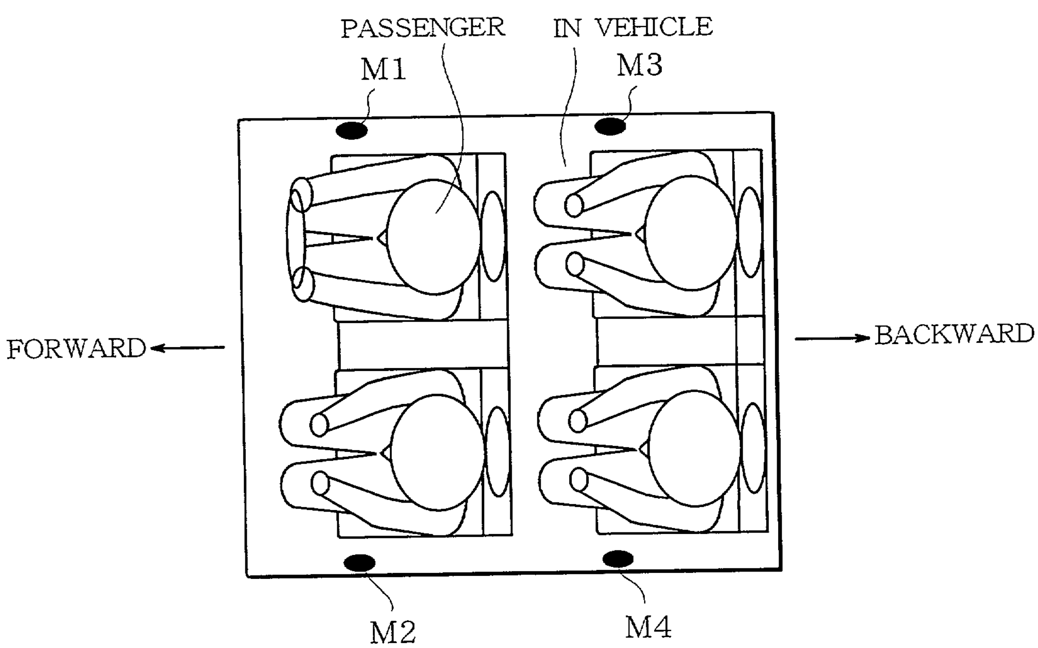
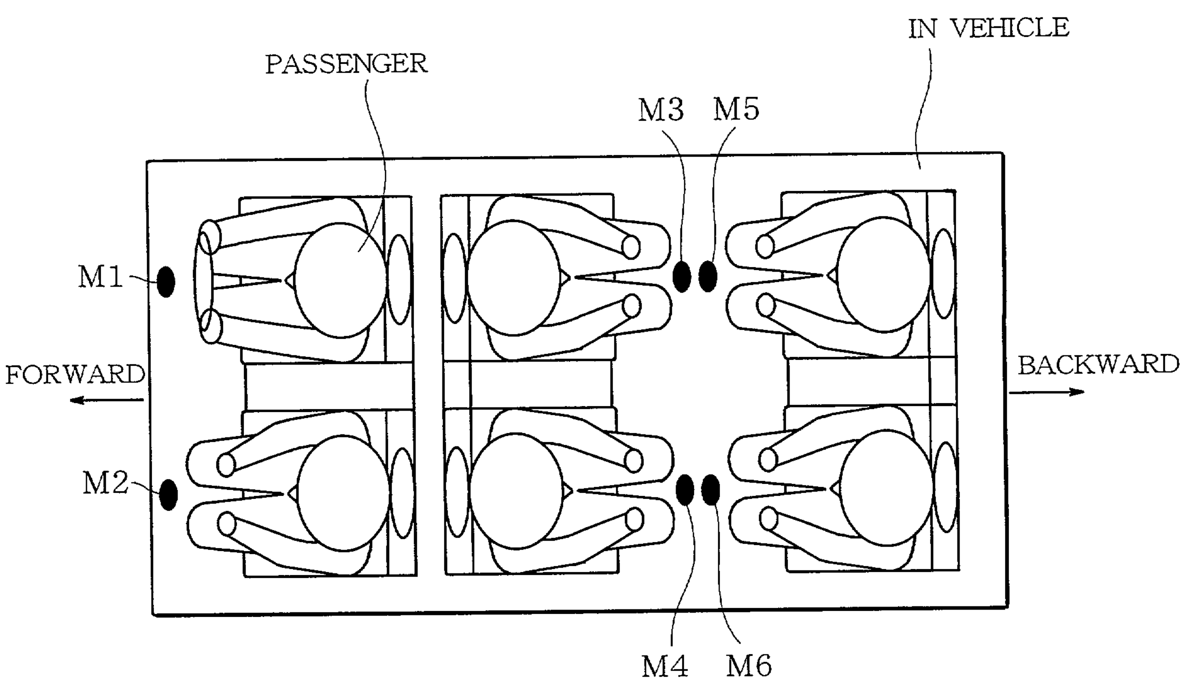

IN VEHICLE
PASSENGER
M3
M1
FORWARD
BACKWARD
M2
M4

FIG. 2 B

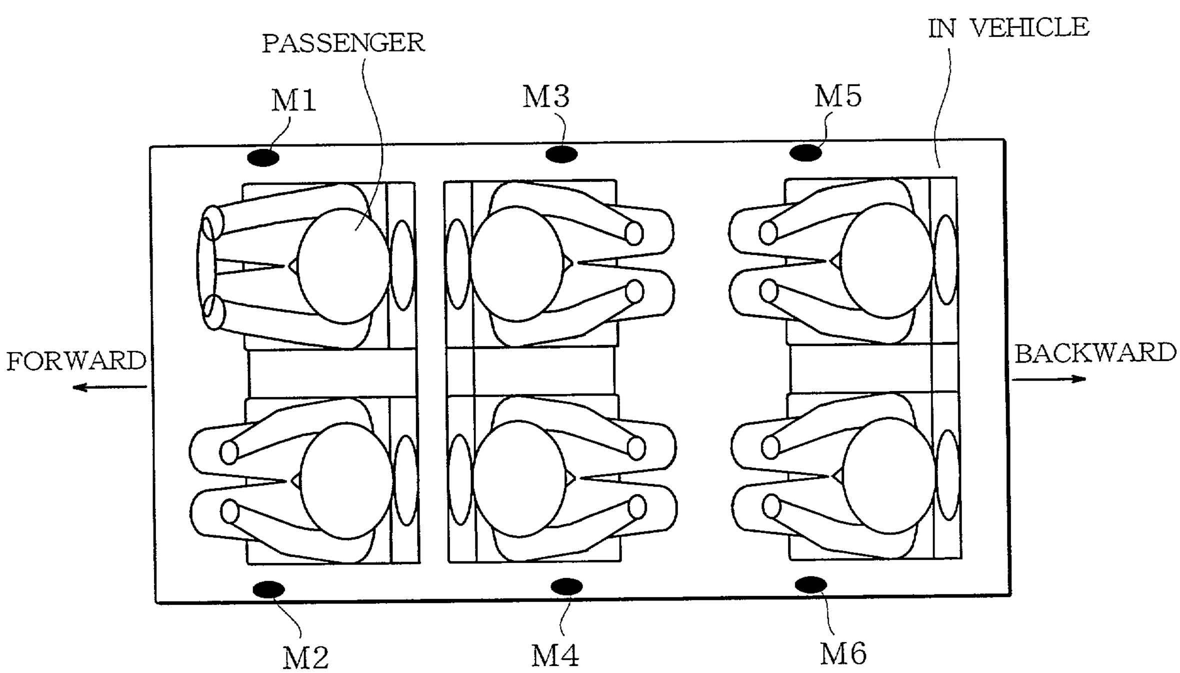

PASSENGER
IN VEHICLE
M1
M3
FORWARD
BACKWARD
M2
M4

F I G .3 A



IN VEHICLE
PASSENGER
M3
M5
M1
FORWARD
BACKWARD
M2
M4
M6

F I G .3 B



IN VEHICLE
PASSENGER
M1
M3
M5
FORWARD
BACKWARD
M2
M4
M6

F I G .4

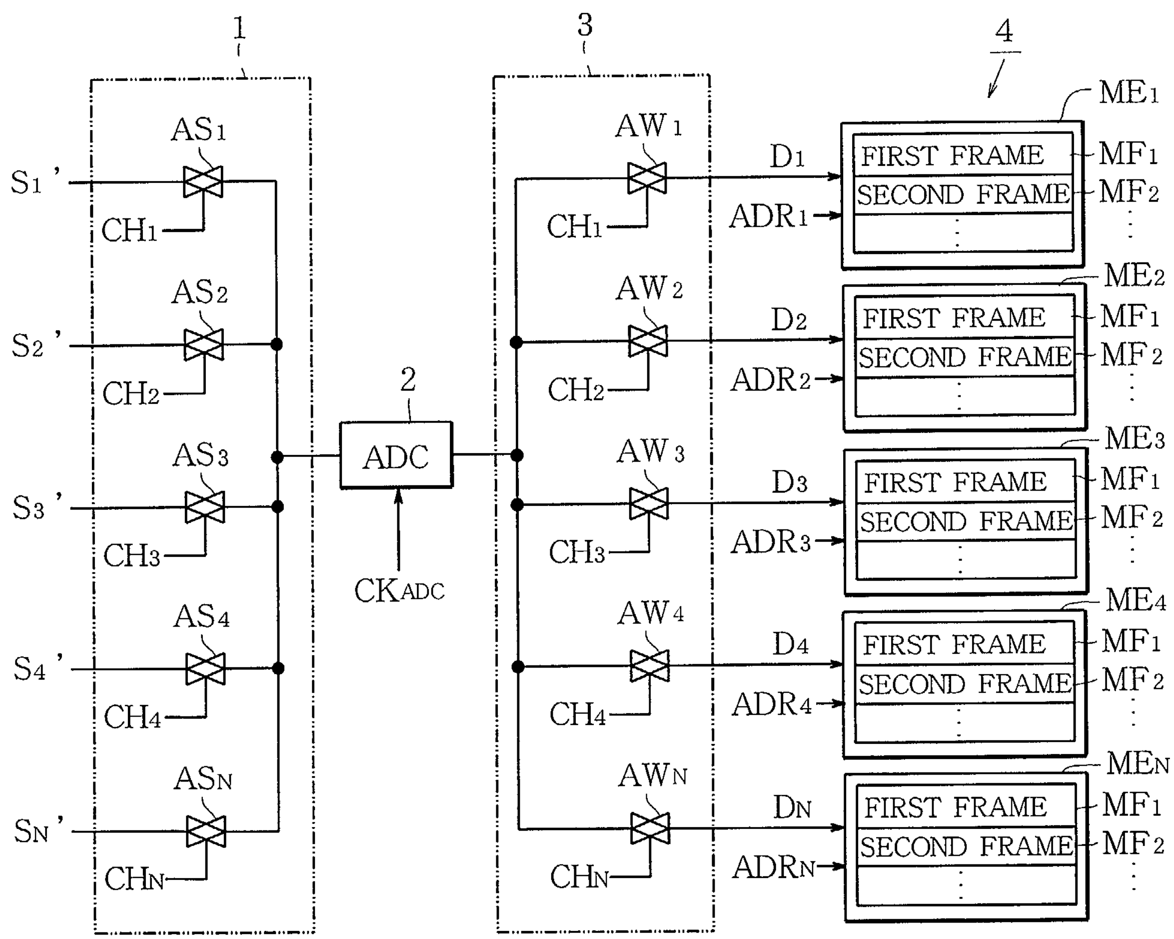

1
2
3
4
S1'
S2'
S3'
S4'
SN'
AS1
AS2
AS3
AS4
ASN
CH1
CH2
CH3
CH4
CHN
ADC
CKADC
AW1
AW2
AW3
AW4
AWN
D1
D2
D3
D4
DN
ADR1
ADR2
ADR3
ADR4
ADRN
ME1
ME2
ME3
ME4
MEN
MF1
MF2
FIRST FRAME
SECOND FRAME

FIG. 5

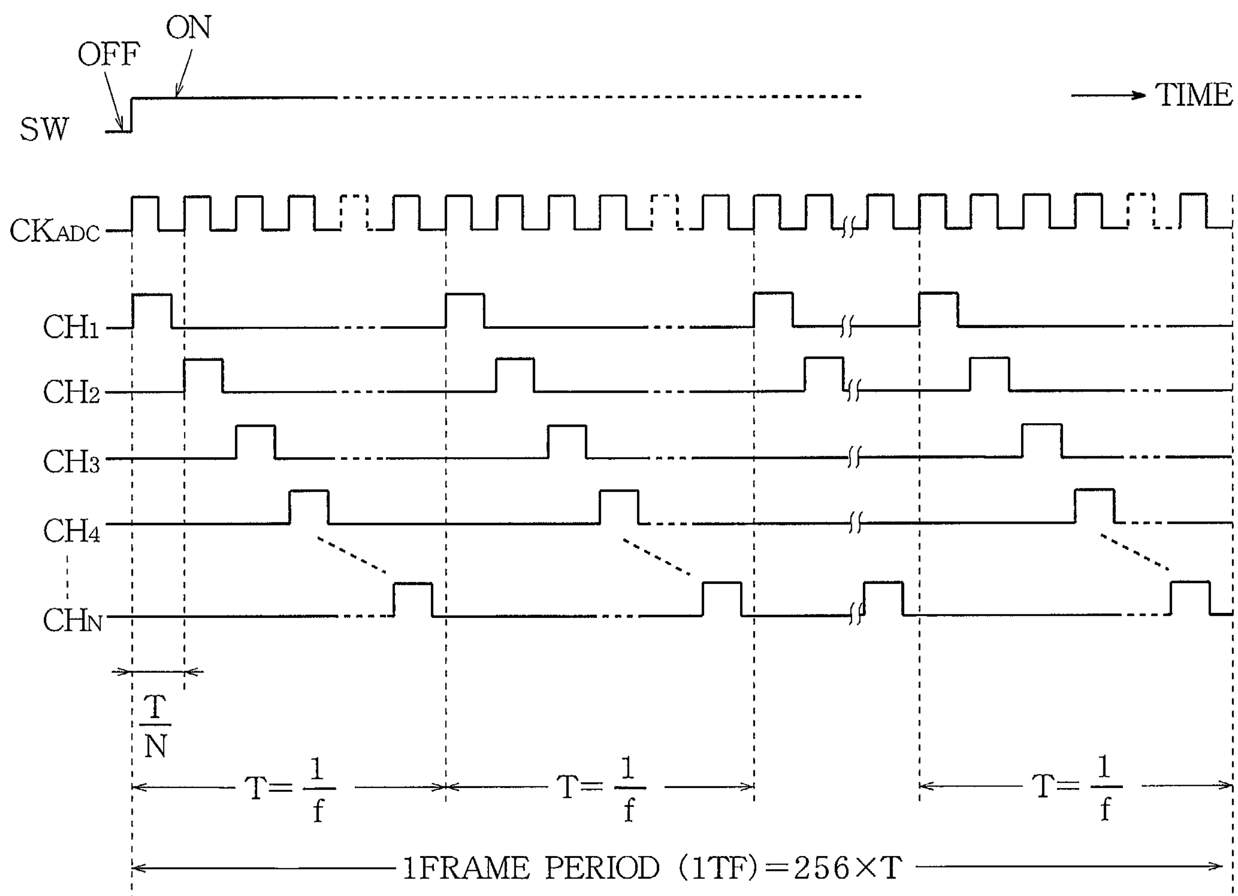

ON
OFF
SW
TIME
CKADC
CH1
CH2
CH3
CH4
CHN
T/N
T= 1/f
T= 1/f
T= 1/f
1FRAME PERIOD (1TF)=256×T

FIG. 6 A

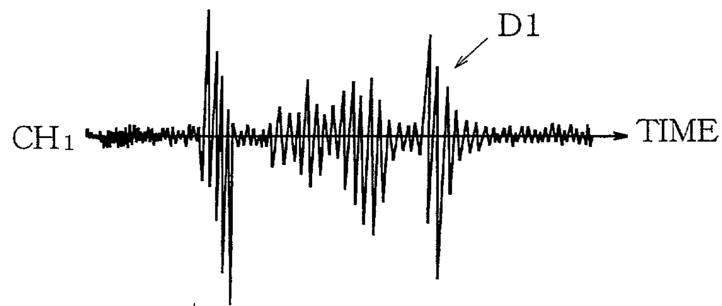


FIG. 6 B

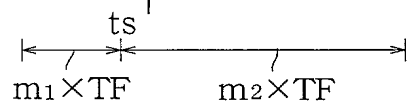


FIG. 6 C

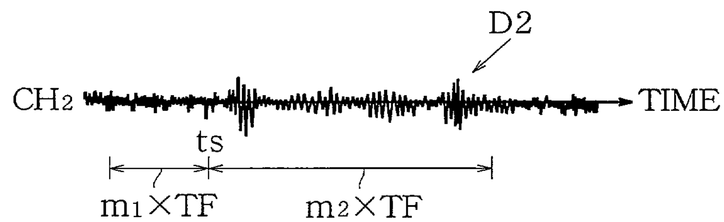


FIG. 6 D

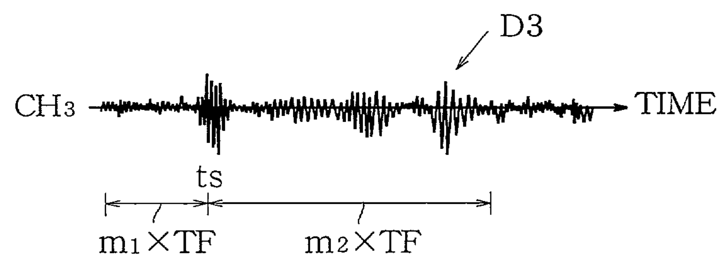
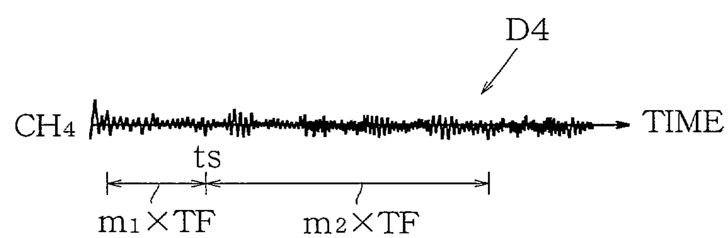


FIG. 6 E



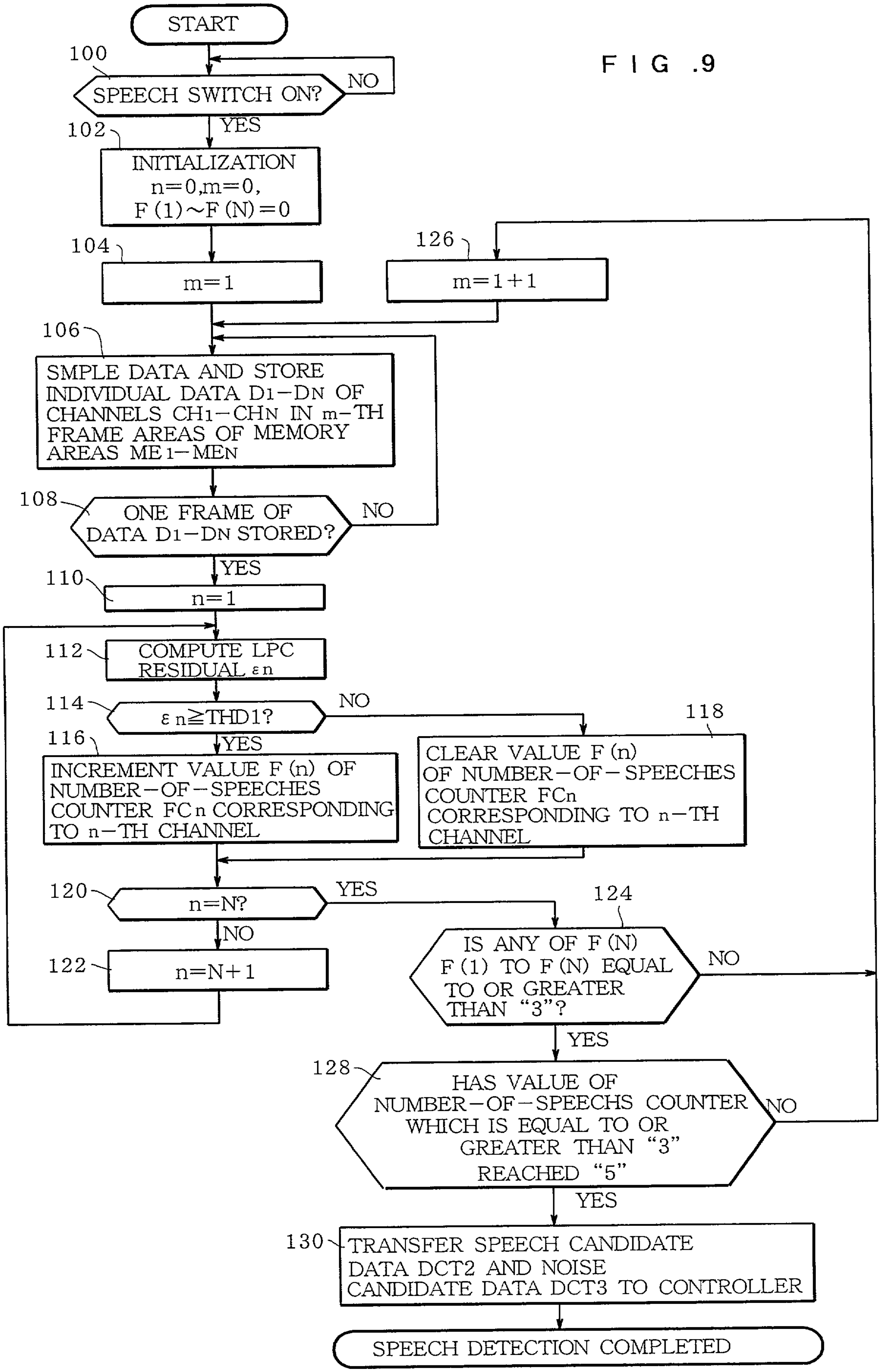
F I G .7

RANKING	AVERAGE S/N	AVERAGE VOICE POWER
Rnk0	MAXIMUM VALUE	MAXIMUM VALUE
Rnk1	THD2 OR GREATER	THD3 OR GREATER
Rnk2	THD2 OR GREATER	SMALLER THAN THD3
Rnk3	SMALLER THAN THD2	THD3 OR GREATER
Rnk4	SMALLER THAN THD2	SMALLER THAN THD3

F I G .8

	CH_1	CH_2	CH_3	CH_4	·····	NOISE CHANNEL
CASE 1	Rnk0	Rnk1	Rnk2	Rnk1	·····	CH_3
CASE 2	Rnk0	Rnk1	Rnk1	Rnk2	·····	CH_4
CASE 3	Rnk0	Rnk1	Rnk1	Rnk3	·····	CH_4
CASE 4	Rnk0	Rnk1	Rnk1	Rnk4	·····	CH_4
CASE 5	Rnk0	Rnk1	Rnk2	Rnk1	·····	CH_3
CASE 6	Rnk0	Rnk1	Rnk2	Rnk2	·····	CH_3
CASE 7	Rnk0	Rnk1	Rnk2	Rnk3	·····	CH_3
CASE 8	Rnk0	Rnk1	Rnk2	Rnk4	·····	CH_3
CASE 9	Rnk0	Rnk1	Rnk3	Rnk1	·····	CH_3

F I G .9



START
100
SPEECH SWITCH ON?
NO
YES
102
INITIALIZATION n=0,m=0, F (1)~F (N)=0
104
m=1
126
m=1+1
106
SMPLE DATA AND STORE INDIVIDUAL DATA D1-DN OF CHANNELS CH1-CHN IN m-TH FRAME AREAS OF MEMORY AREAS ME1-MEN
108
ONE FRAME OF DATA D1-DN STORED?
NO
YES
110
n=1
112
COMPUTE LPC RESIDUAL εn
114
εn≧THD1?
NO
118
YES
116
INCREMENT VALUE F (n) OF NUMBER-OF-SPEECHES COUNTER FCn CORRESPONDING TO n-TH CHANNEL
CLEAR VALUE F (n) OF NUMBER-OF-SPEECHES COUNTER FCn CORRESPONDING TO n-TH CHANNEL
120
n=N?
YES
124
NO
122
n=N+1
IS ANY OF F (N) F (1) TO F (N) EQUAL TO OR GREATER THAN "3"?
NO
YES
128
HAS VALUE OF NUMBER-OF-SPEECHS COUNTER WHICH IS EQUAL TO OR GREATER THAN "3" REACHED "5"
NO
YES
130
TRANSFER SPEECH CANDIDATE DATA DCT2 AND NOISE CANDIDATE DATA DCT3 TO CONTROLLER
SPEECH DETECTION COMPLETED

F I G .1 0

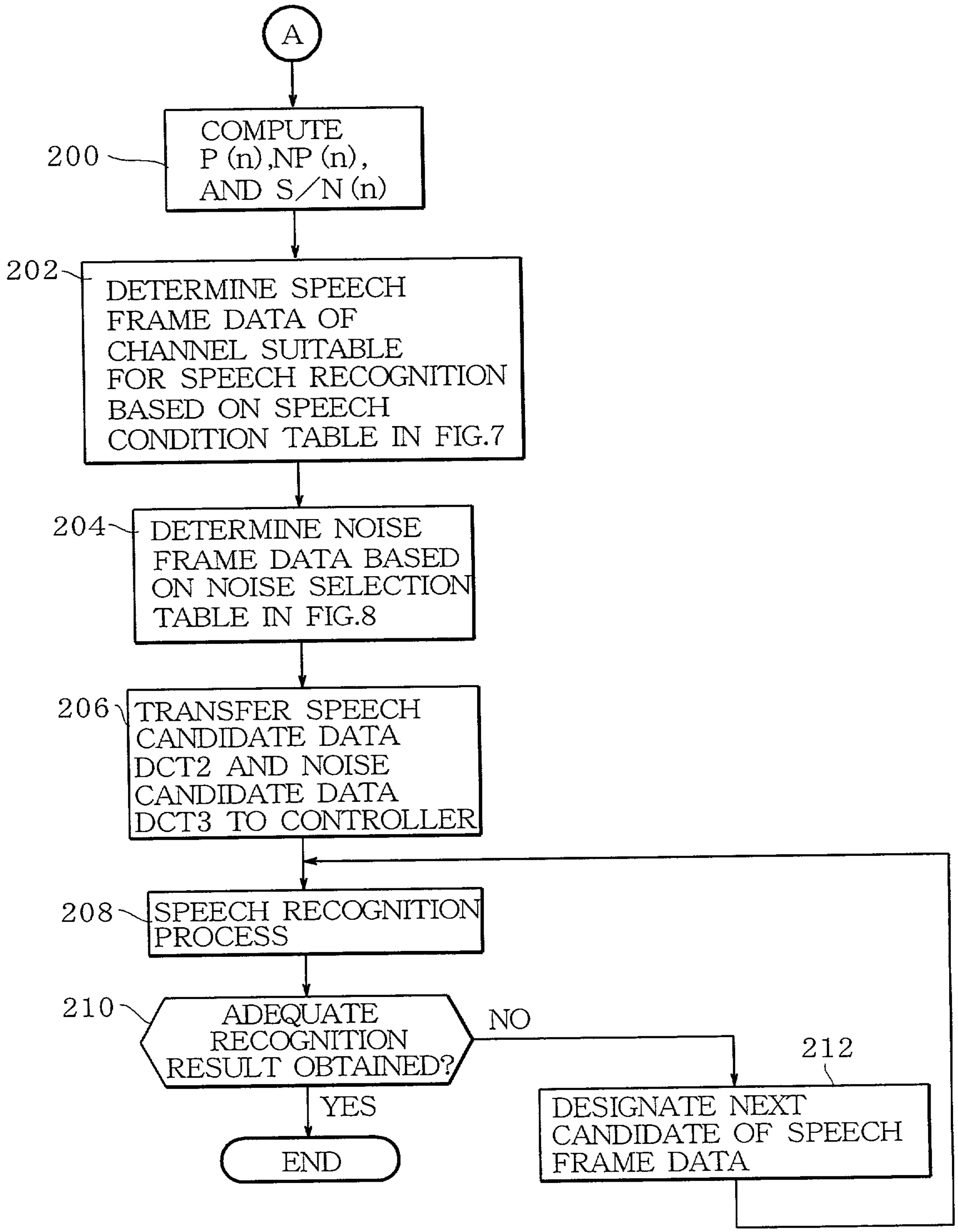
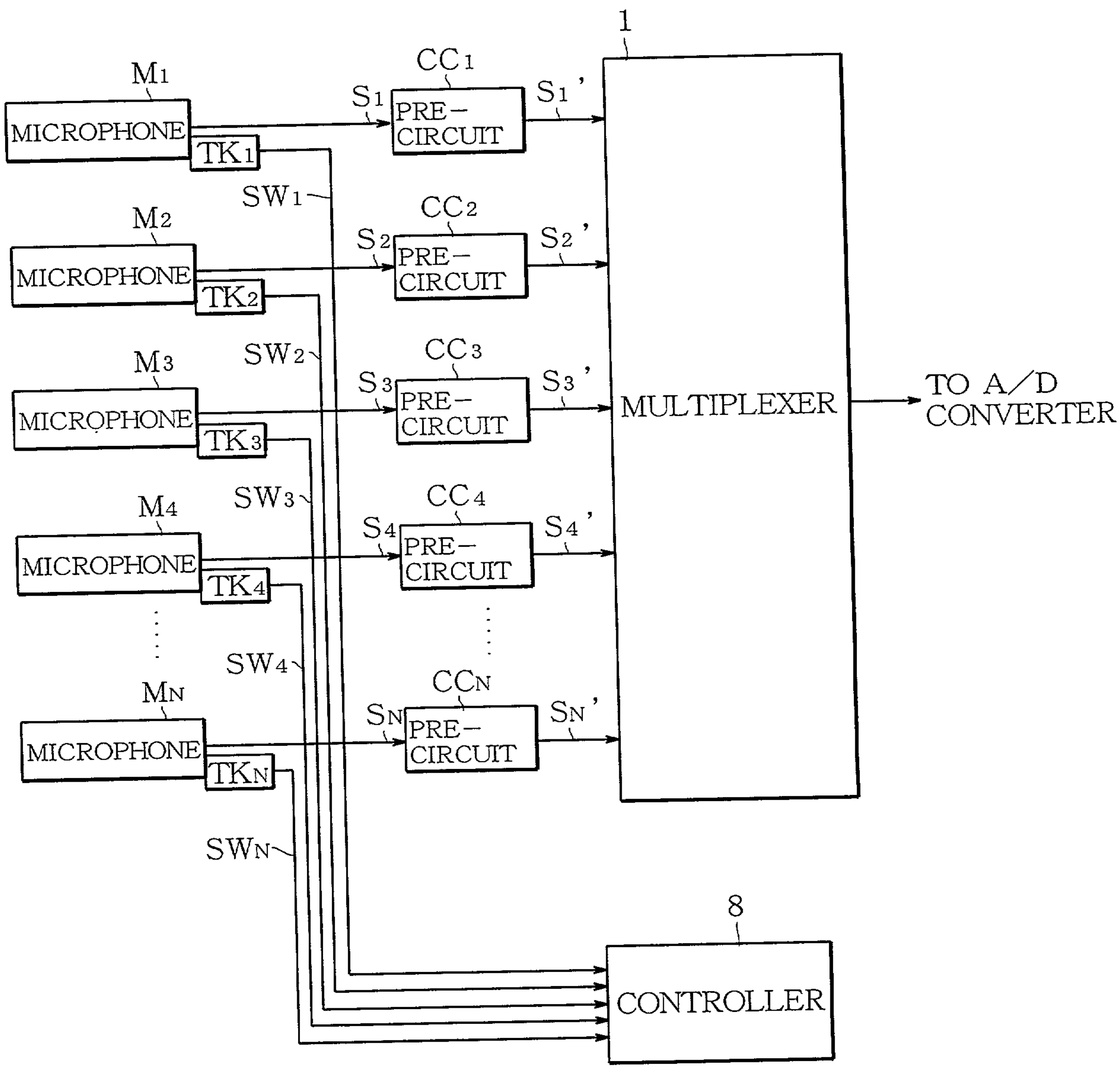

A
200
COMPUTE
P (n),NP (n),
AND S/N (n)
202
DETERMINE SPEECH FRAME DATA OF CHANNEL SUITABLE FOR SPEECH RECOGNITION BASED ON SPEECH CONDITION TABLE IN FIG.7
204
DETERMINE NOISE FRAME DATA BASED ON NOISE SELECTION TABLE IN FIG.8
206
TRANSFER SPEECH CANDIDATE DATA DCT2 AND NOISE CANDIDATE DATA DCT3 TO CONTROLLER
208
SPEECH RECOGNITION PROCESS
210
ADEQUATE RECOGNITION RESULT OBTAINED?
NO
YES
212
DESIGNATE NEXT CANDIDATE OF SPEECH FRAME DATA
END

FIG. 11


1
MICROPHONE
MICROPHONE
MICROPHONE
MICROPHONE
MICROPHONE
TK1
TK2
TK3
TK4
TKN
M1
M2
M3
M4
MN
SW1
SW2
SW3
SW4
SWN
CC1
CC2
CC3
CC4
CCN
PRE-CIRCUIT
PRE-CIRCUIT
PRE-CIRCUIT
PRE-CIRCUIT
PRE-CIRCUIT
S1
S2
S3
S4
SN
S1'
S2'
S3'
S4'
SN'
MULTIPLEXER
TO A/D CONVERTER
8
CONTROLLER

N, M, M & O Docket No. ____________ NIKAIDO, MARMELSTEIN, MURRAY & ORAM LLP

Declaration For U.S. Patent Application

As a below named inventor, I hereby declare that:
My residence, post office address and citizenship are as stated below my name.
I believe I am the original, first and sole inventor (if only one name is listed below) or an original, first and joint inventor (if plural names are listed below) of the subject matter which is claimed and for which a patent is sought on the invention entitled (Insert Title) "Speech Recognition System and Method"
the specification of which is attached hereto unless the following box is checked:

☐ was filed on ____________ as United States Application Number or PCT International Application Number ____________ and was amended on ____________ (if applicable).

I hereby state that I have reviewed and understand the contents of the above-identified specification, including the claim(s), as amended by any amendment referred to above.
I acknowledge the duty to disclose information which is material to patentability as defined in 37 C.F.R. §1.56.
I hereby claim foreign priority benefits under 35 U.S.C. §119(a)-(d) or §365(b) of any foreign application(s) for patent or inventor's certificate, or §365(a) of any PCT International application which designated at least one country other than the United States, listed below and have also identified below any foreign application for patent or inventor's certificate or PCT International Application having a filing date before that of the application(s) for which priority is claimed:

(List prior foreign applications. See note A on back of this page)

(Number)	(Country)	(Day/Month/Year Filed)	Priority Claimed
11-246393	Japan	31/08/1999	☒ Yes ☐ No
			☐ Yes ☐ No
			☐ Yes ☐ No

I hereby claim the benefit under 35 U.S.C. §119(e) of any United States provisional application(s) listed below.

(Application Number)	(Filing Date)

(See Note B on back of this page) ☐ See attached list for additional prior foreign or provisional applications.

I hereby claim the benefit under 35 U.S.C. §120 of any United States application(s) or §365(c) of any PCT International application(s) designating the United States of America listed below and, insofar as the subject matter of each of the claims of this application is not disclosed in the prior application(s) (U.S. or PCT) in the manner provided by the first paragraph of 35, U.S.C. §112, I acknowledge the duty to disclose information which is material to patentability as defined in 37 C.F.R. §1.56 which became available between the filing date of the prior application and the national or PCT International filing date of this application.

(List prior U.S. Applications or PCT International applications designating the U.S.)

(Application Serial No.)	(Filing Date)	(Status) (patented, pending, abandoned)

And I hereby appoint as principal attorneys David T. Nikaido, Reg. No. 22,663; Charles M. Marmelstein, Reg. No. 25,895; George E. Oram, Jr., Reg. No. 27,931; Robert B. Murray, Reg. No. 22,980; Martin S. Postman, Reg. No. 18,570; E. Marcie Emas, Reg. No. 32,131; Douglas H. Goldhush, Reg. No. 33,125; Kevin C. Brown, Reg. No. 32,402; Monica Chin Kitts, Reg. No. 36,105; and Richard J. Berman, Reg. No. 39,107.

Please direct all communications to the following address:
NIKAIDO, MARMELSTEIN, MURRAY & ORAM LLP
Metropolitan Square
655 Fifteenth Street, N.W., Suite 330 - G Street Lobby
Washington, D.C. 20005-5701
(202) 638-5000 Fax: (202) 638-4810

I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true; and further, that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful false statements may jeopardize the validity of the application or any patent issued thereon.

(See Note C on back of this page)

Full name of sole or first inventor Shoutarou YODA
Inventor's signature Shoutarou yoda   August 25, 2000 Date
Residence Saitama-ken, Japan
Citizenship Japan
Post Office Address c/o Kawagoe Koujou, Pioneer Corporation, 25-1 Aza Nishimachi, Yamada, Kawagoe-shi, Saitama-ken 350-8555 Japan

IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re application of:

Shoutarou YODA

Serial No.: New Application

Filed: August 30, 2000

For: SPEECH RECOGNITION SYSTEM AND METHOD

NOTIFICATION OF CHANGE OF NAME AND ADDRESS

Commissioner for Patents
Washington, D. C. 20231

August 30, 2000

Sir:

It is respectfully requested that the correspondence address for the above-identified application be changed to the following:

ARENT FOX KINTNER PLOTKIN & KAHN, PLLC
1050 Connecticut Avenue, N.W.
Suite 600
Washington, D. C. 20036-5339
Tel: (202) 857-6000
Fax: (202) 638-4810

In the event that any fees are due with respect to this paper, please charge our Deposit Account No. 01-2300.

Respectfully submitted,

ARENT FOX KINTNER PLOTKIN & KAHN, PLLC

David T. Nikaido
Reg. No. 22,663

Atty. Docket No.: P107156-00019

1050 Connecticut Avenue, N.W.
Suite 600
Washington, D. C. 20036-5339
Tel: (202) 857-6000
Fax: (202) 638-4810

DTN:mso